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(54) Abstract Title
Telecommunication services identification

(57) A method and apparatus are disclosed for identifying telecommunications services. The apparatus enables call establishment between a terminal of an originating network and a terminal of a terminating network. It comprises an input for receiving call type information in a first format from the originating network, means for reformatting received call type information into a second format, output means for outputting the call type information in the second format over the terminating network; and connection means for completing an appropriate connection between the terminals.

At least one drawing originally filed was informal and the print reproduced here is taken from a later filed formal copy.

This print takes account of replacement documents submitted after the date of filing to enable the application to comply with the formal requirements of the Patents Rules 1995

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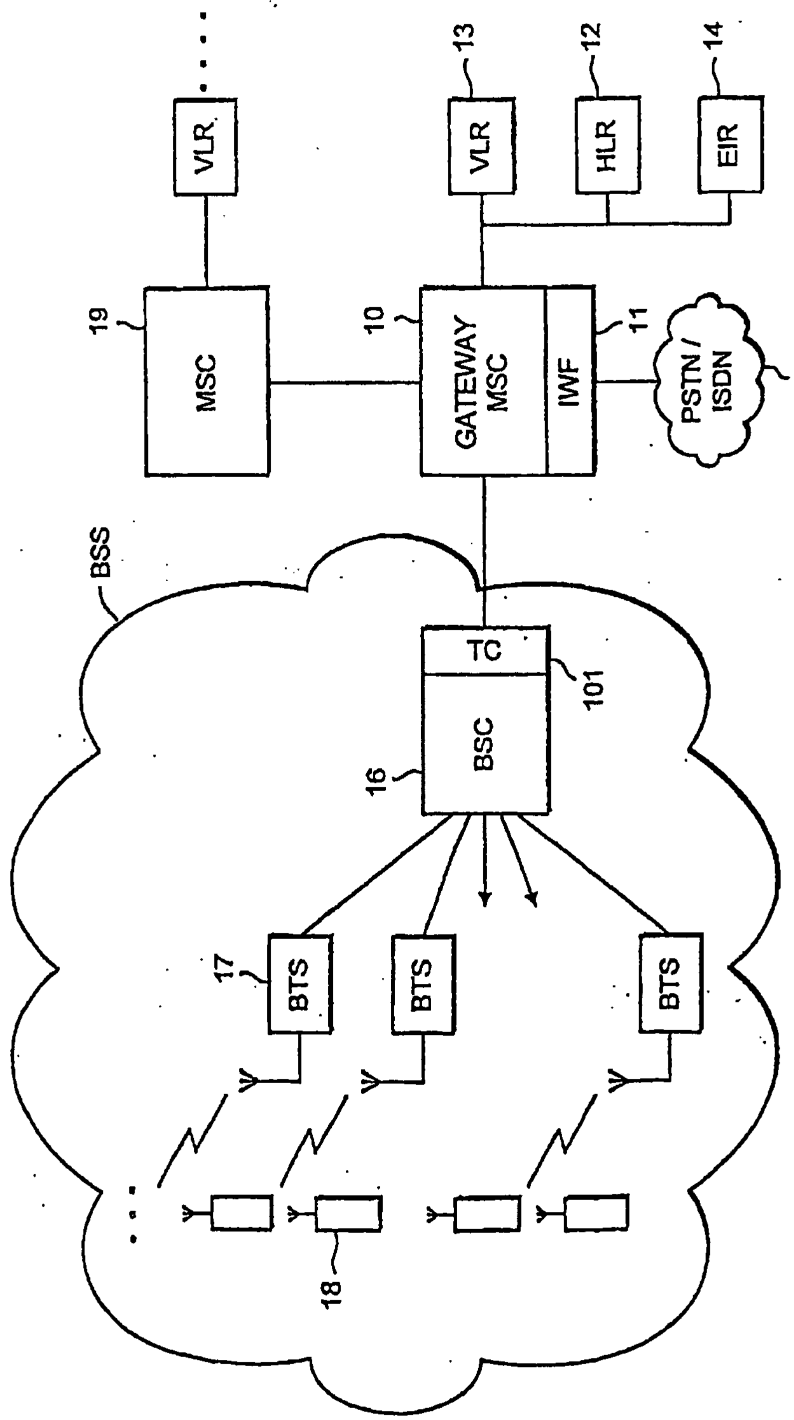
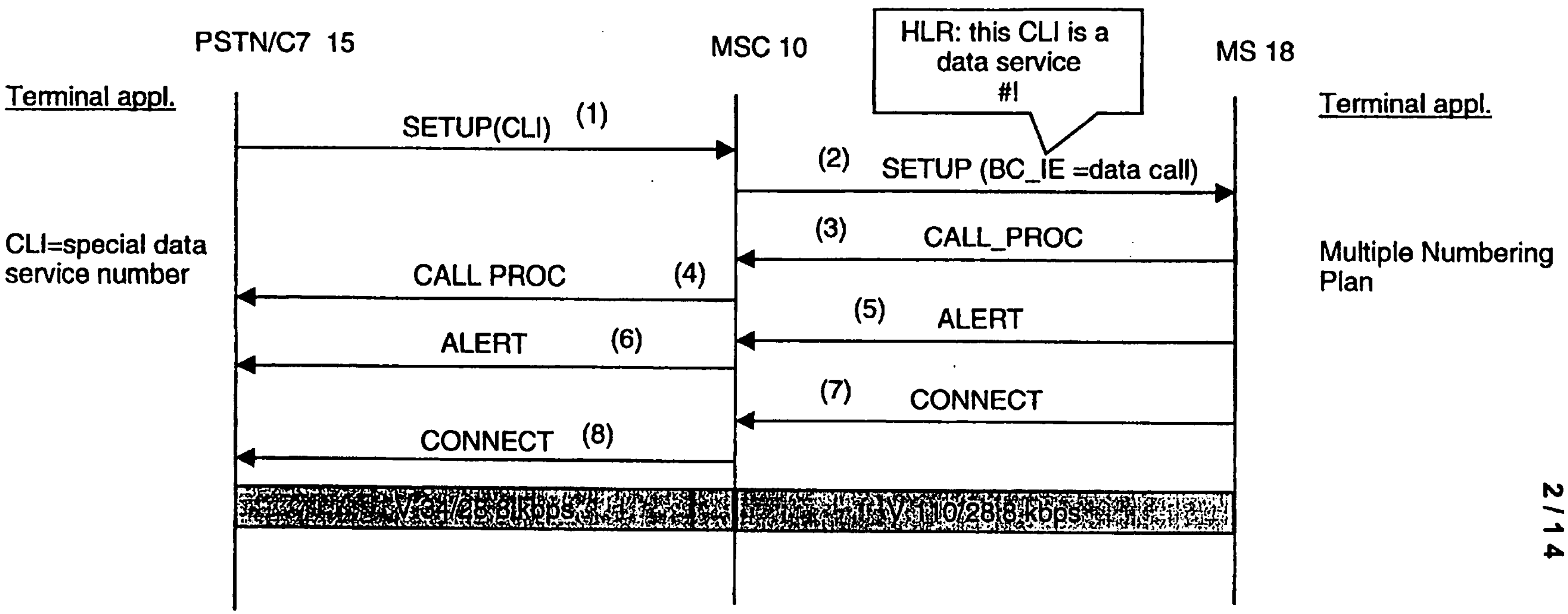

VLR
MSC
19
VLR
13
HLR
12
EIR
14
GATEWAY MSC
10
IWF
11
PSTN / ISDN
15
BSS
TC
101
BSC
16
BTS
17
BTS
BTS
18

FIGURE 1



PSTN/C7 15
MSC 10
MS 18
HLR: this CLI is a data service #!
Terminal appl.
Terminal appl.
SETUP(CLI) (1)
(2) SETUP (BC_IE =data call)
(3) CALL_PROC
CLI=special data service number
Multiple Numbering Plan
CALL PROC (4)
(5) ALERT
ALERT (6)
(7) CONNECT
CONNECT (8)

FIGURE 2a

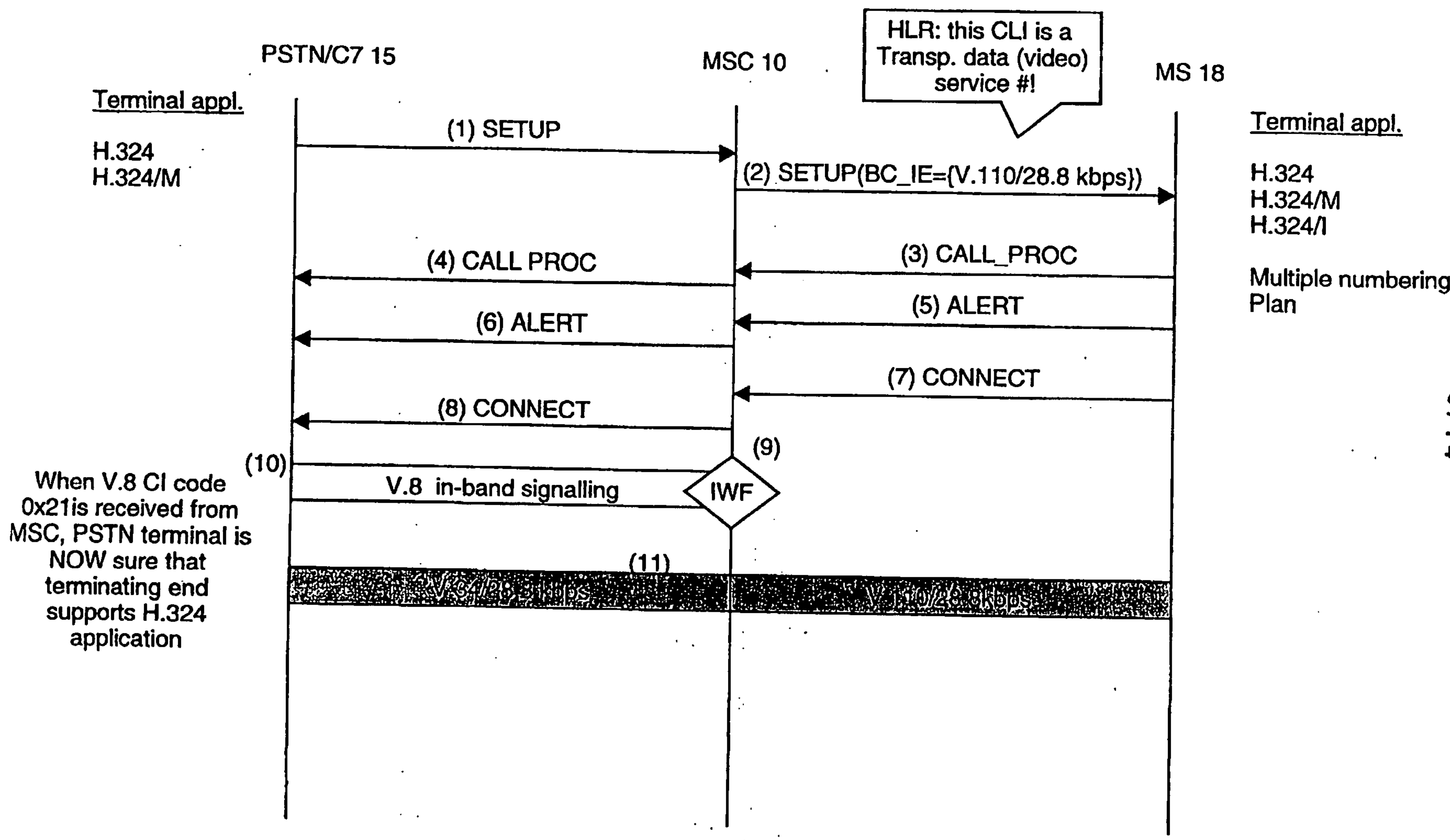

PSTN/C7 15
MSC 10
MS 18
HLR: this CLI is a Transp. data (video) service #!
Terminal appl.
H.324
H.324/M
Terminal appl.
H.324
H.324/M
H.324/I
Multiple numbering Plan
(1) SETUP
(2) SETUP(BC_IE={V.110/28.8 kbps})
(3) CALL_PROC
(4) CALL PROC
(5) ALERT
(6) ALERT
(7) CONNECT
(8) CONNECT
(9)
IWF
(10)
V.8 in-band signalling
When V.8 CI code 0x21is received from MSC, PSTN terminal is NOW sure that terminating end supports H.324 application
(11)
V.34/28.8 kbps
V.110/28.8kbps

FIGURE 2b

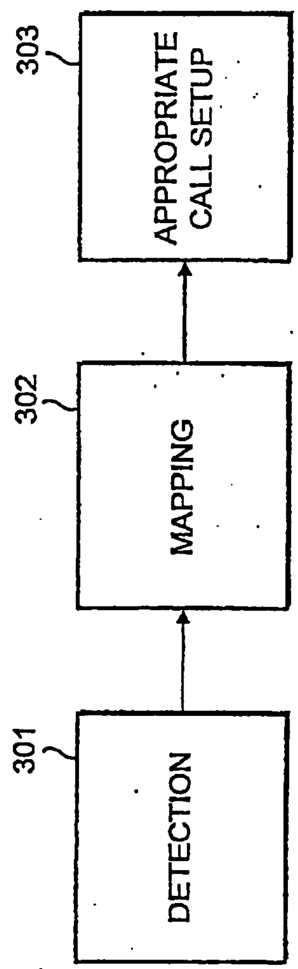


FIGURE 3

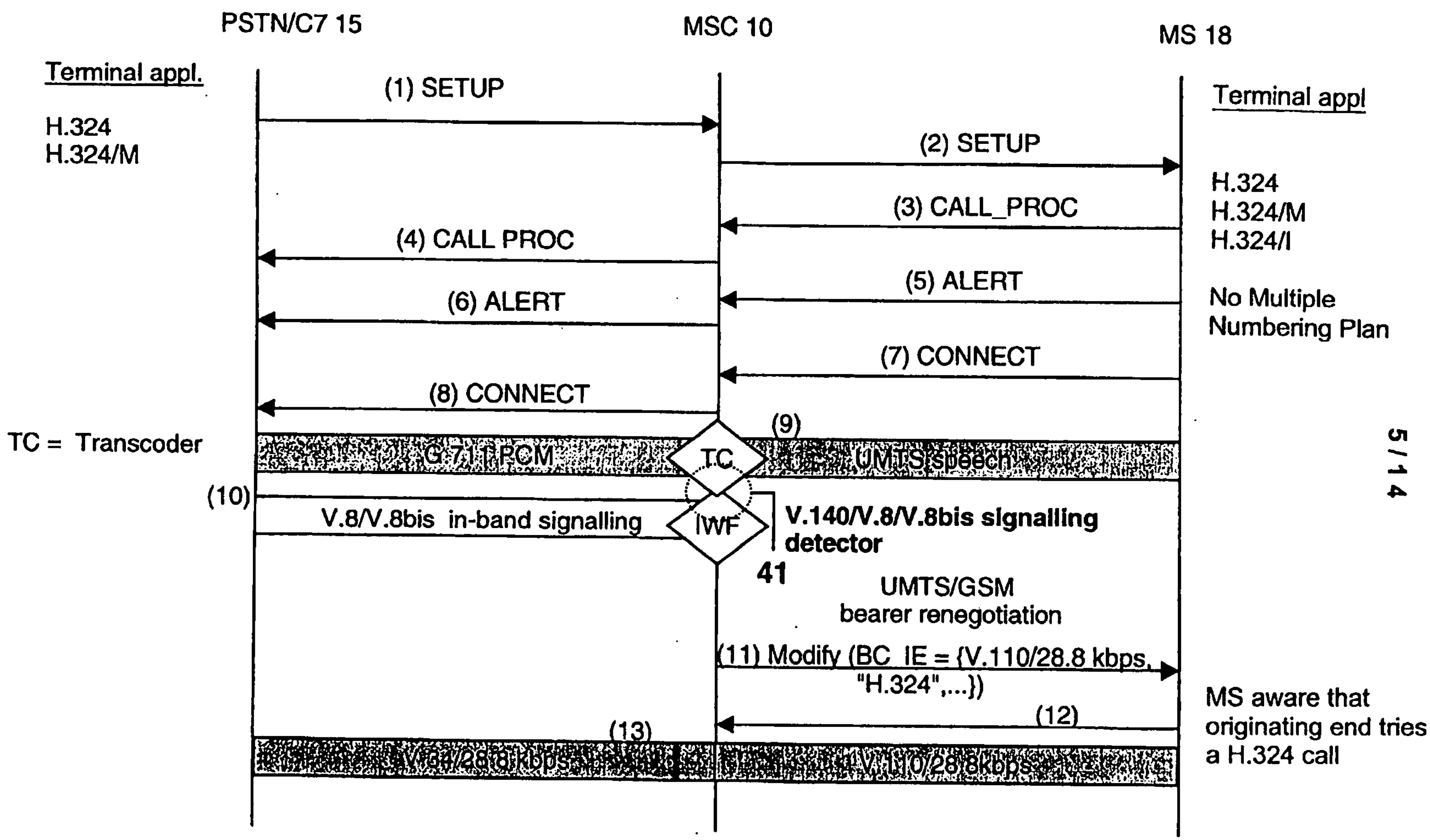

PSTN/C7 15
MSC 10
MS 18
Terminal appl.
H.324
H.324/M
Terminal appl
H.324
H.324/M
H.324/I
No Multiple
Numbering Plan
(1) SETUP
(2) SETUP
(3) CALL_PROC
(4) CALL PROC
(5) ALERT
(6) ALERT
(7) CONNECT
(8) CONNECT
(9)
TC = Transcoder
G.711 PCM
TC
UMTS speech
(10)
V.8/V.8bis in-band signalling
IWF
V.140/V.8/V.8bis signalling
detector
41
UMTS/GSM
bearer renegotiation
(11) Modify (BC_IE = {V.110/28.8 kbps,
"H.324",...})
MS aware that
originating end tries
a H.324 call
(12)
(13)
V.110/28.8kbps

FIGURE 4a

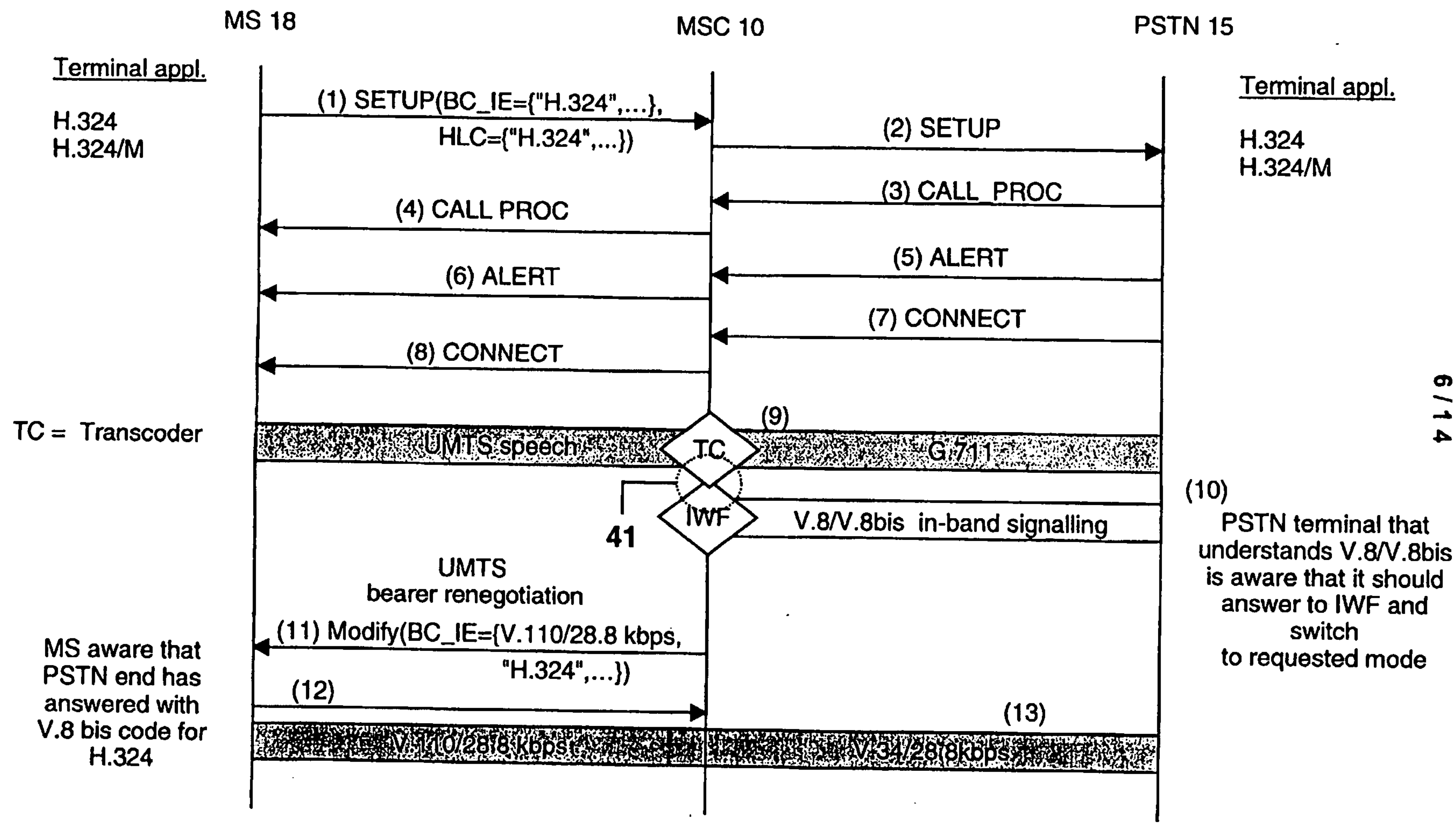

MS 18
MSC 10
PSTN 15
Terminal appl.
H.324
H.324/M
(1) SETUP(BC_IE={"H.324",...},
HLC={"H.324",...})
(2) SETUP
Terminal appl.
H.324
H.324/M
(3) CALL_PROC
(4) CALL PROC
(5) ALERT
(6) ALERT
(7) CONNECT
(8) CONNECT
TC = Transcoder
(9)
UMTS speech
TC
G.711
(10)
IWF
41
V.8/V.8bis in-band signalling
PSTN terminal that understands V.8/V.8bis is aware that it should answer to IWF and switch to requested mode
UMTS
bearer renegotiation
(11) Modify(BC_IE={V.110/28.8 kbps,
"H.324",...})
MS aware that PSTN end has answered with V.8 bis code for H.324
(12)
(13)
V.110/28.8kbps
V.34/28.8kbps

FIGURE 4b

MS - PSTN direct call without V.8bis involvement

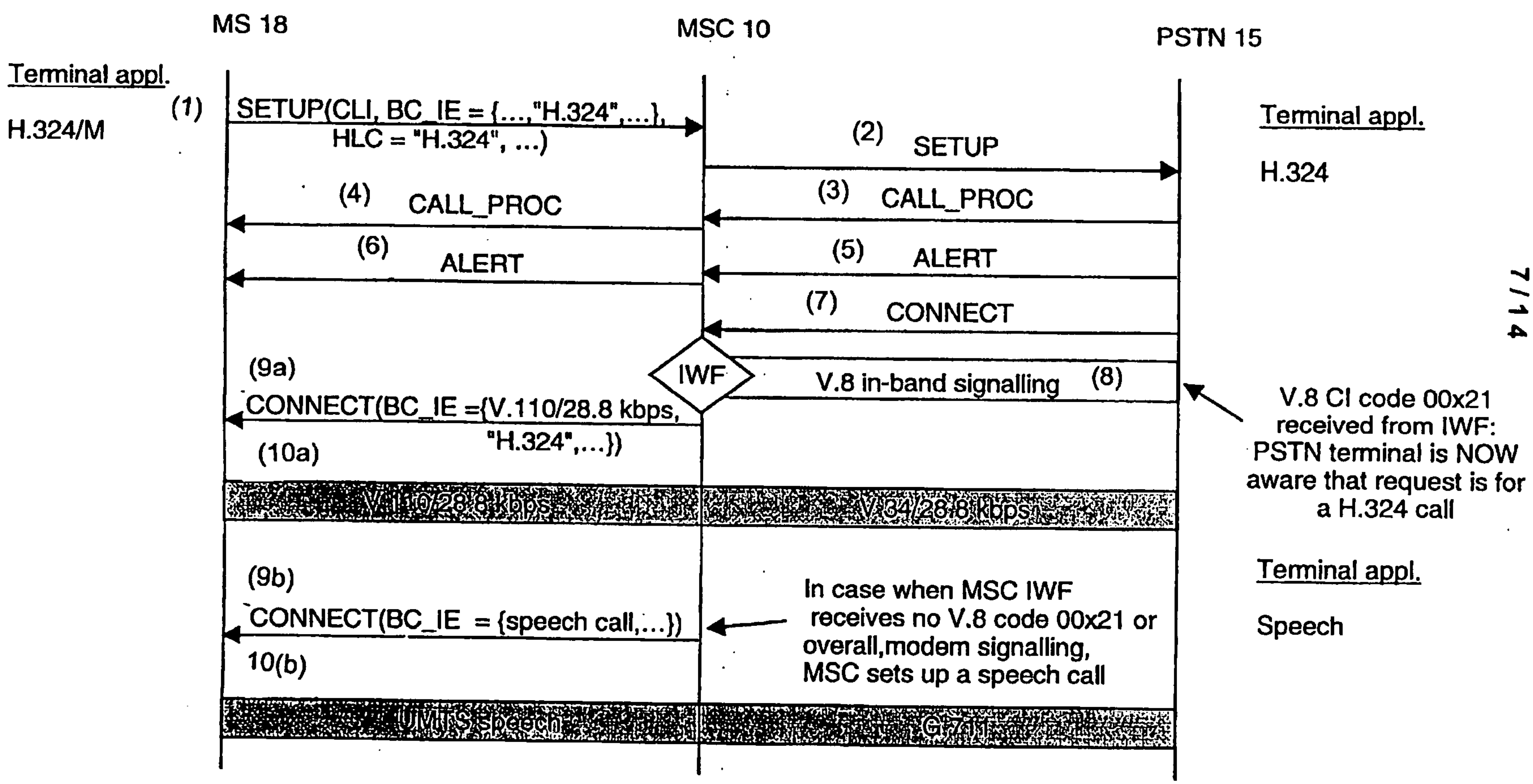
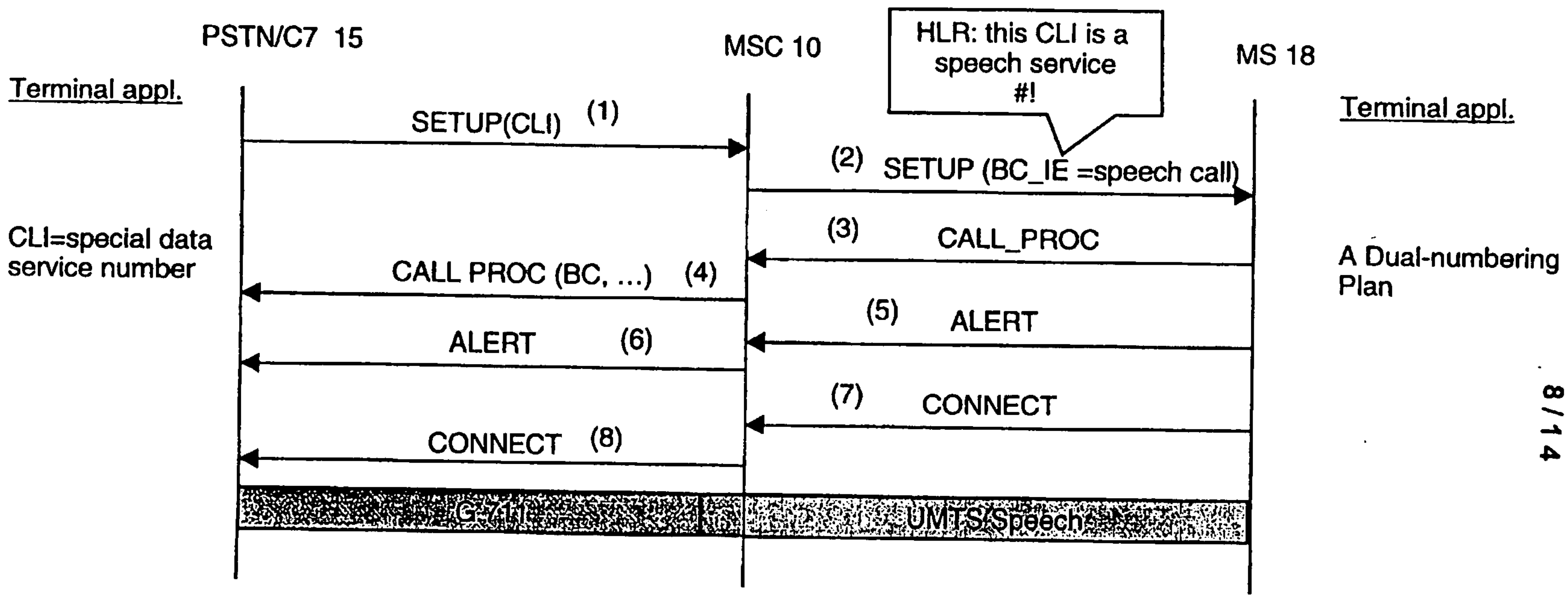
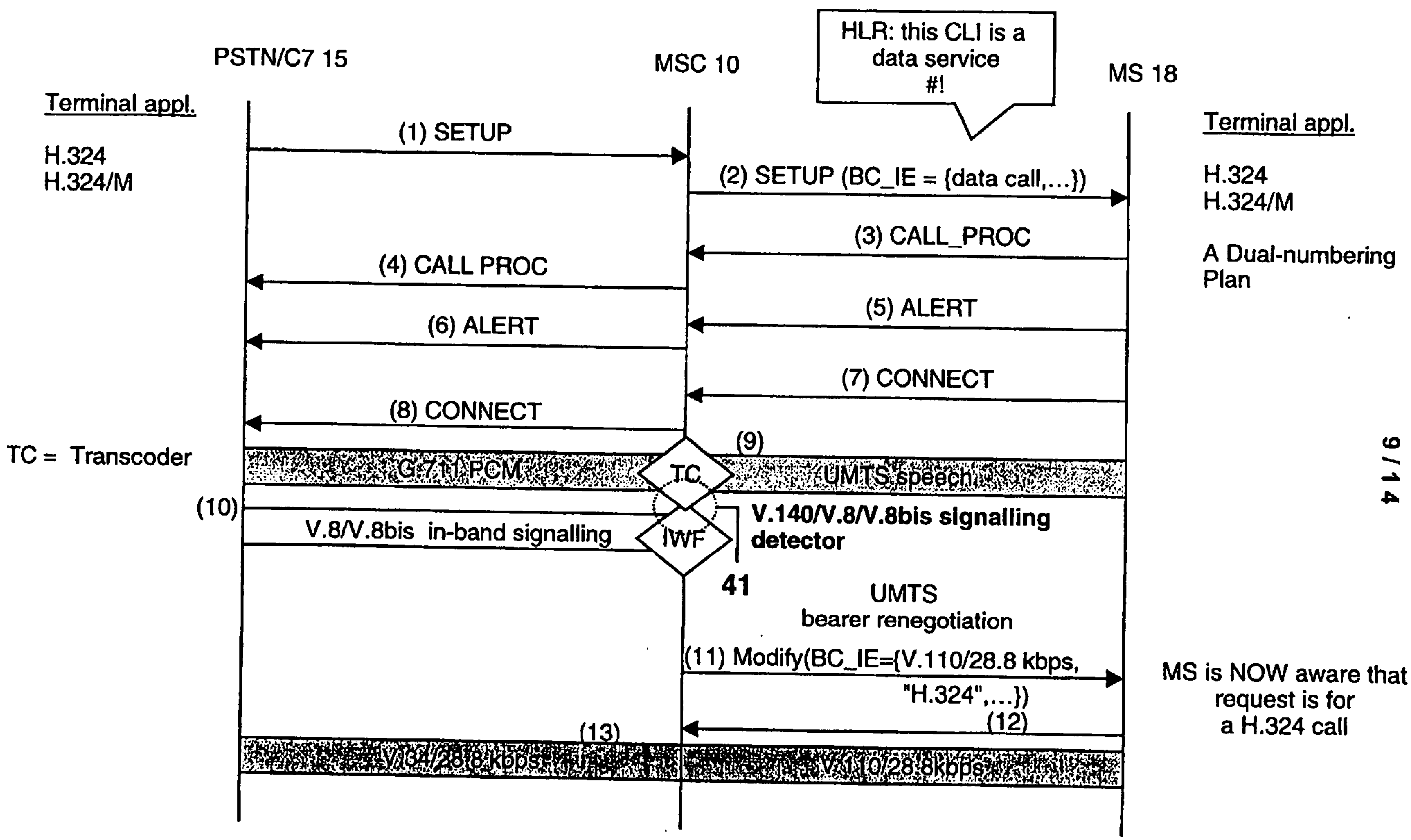


FIGURE 4c



PSTN/C7 15
MSC 10
MS 18
Terminal appl.
Terminal appl.
HLR: this CLI is a speech service #!
SETUP(CLI) (1)
(2) SETUP (BC_IE =speech call)
(3) CALL_PROC
CALL PROC (BC, ...) (4)
(5) ALERT
ALERT (6)
(7) CONNECT
CONNECT (8)
G.711
UMTS/Speech
CLI=special data service number
A Dual-numbering Plan

FIGURE 5a



PSTN/C7 15
MSC 10
MS 18
HLR: this CLI is a data service #!
Terminal appl.
H.324
H.324/M
Terminal appl.
H.324
H.324/M
A Dual-numbering Plan
(1) SETUP
(2) SETUP (BC_IE = {data call,...})
(3) CALL_PROC
(4) CALL PROC
(5) ALERT
(6) ALERT
(7) CONNECT
(8) CONNECT
(9)
TC = Transcoder
G.711 PCM
TC
UMTS speech
(10)
V.8/V.8bis in-band signalling
IWF
V.140/V.8/V.8bis signalling detector
41
UMTS bearer renegotiation
(11) Modify(BC_IE={V.110/28.8 kbps, "H.324",...})
(12)
MS is NOW aware that request is for a H.324 call
(13)
V.34/28.8 kbps
V.110/28.8kbps

FIGURE 5b

V.8

Call function category (Octet – "callf0")
100 PSTN Multimedia Terminal (Bits 567)

Modulation modes category (Octet – "modn0")
1 V.34 duplex availability (Bit 6)

V.25ter

Modulation control commands
Modulation reporting control (+MR)
+MCR: V34 +MCR: <carrier>
+MRR: 28800 +MRR: <rate>

V.8bis

Standard information field - {SPar(1)} coding
Standard information field – Data {NPar(2)} coding (Octet 2)
1 Rec. V.34 duplex mode (Bit 5)

Standard information field - {SPar(1)} coding
Standard information field – H.324 multimedia terminal {NPar(2)} coding
1 Video (Bit 1)

Standard information field - {SPar(1)} coding
Standard information field – H.324 multimedia terminal {NPar(2)} coding
1 Audio (Bit 2)

UMTS
Bearer Capability

Bearer Capability Information Element
Transfer mode (octet 3)
0 Circuit mode (Bit 4)

Bearer Capability Information Element

Duplex mode (octet 4)
1 Full duplex (Bit 4)

Bearer Capability Information Element
Synchronous/Asynchronous (octet 6)
0 Synchronous (Bit 1)

Bearer Capability Information Element
Fixed network user rate (octet 6d)
00100 28.8 kbps (Bits 54321)

Bearer Capability Information Element
Acceptable channel codings (octet 6e)
1 TCH/F14.4 acceptable (Bit 7)

Bearer Capability Information Element
Acceptable channel codings (octet 6e)

1 TCH/F9.6 acceptable (Bit 5)

Bearer Capability Information Element
Acceptable channel codings (octet 6e)

1 TCH/F4.8 acceptable (Bit 4)

Bearer Capability Information Element
Maximum number of traffic channels (octet 6e)

001 2 TCH (Bits 321)

Bearer Capability Information Element
Connection element (octet 6c)
00 Transparent (Bits 76)

Bearer Capability Information Element
Other modem type (octet 6d)
10 V.34 (Bits 76)

High Layer Compatibility

High Layer Compatibility Information Element
High layer characteristics identification (octet 4)
1100001 PSTN Multimedia Terminal (Bits 7654321)

Figure 6

ISDN - UMTS

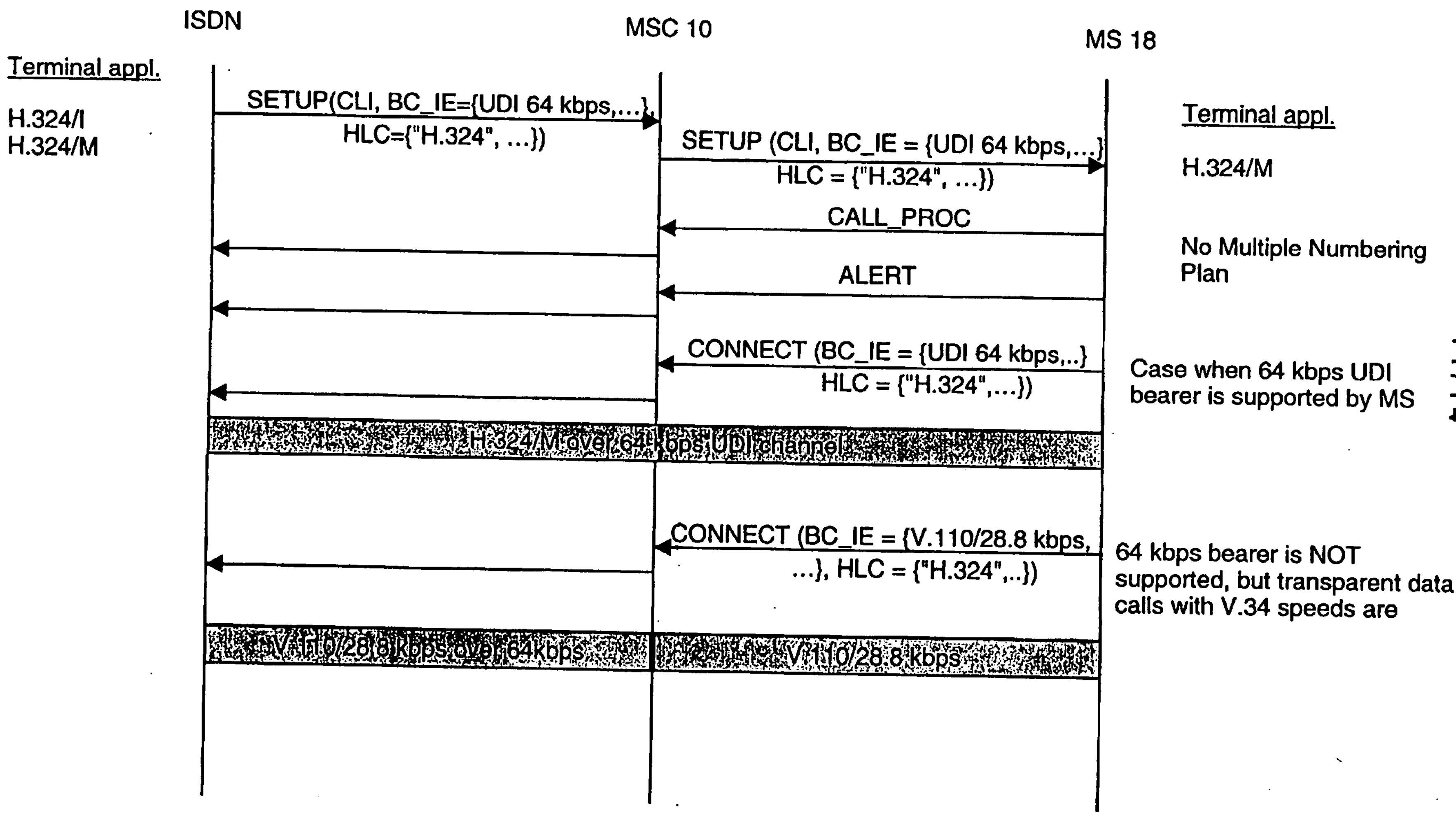


FIGURE 7a

ISDN - UMTS without end-to-end HLC

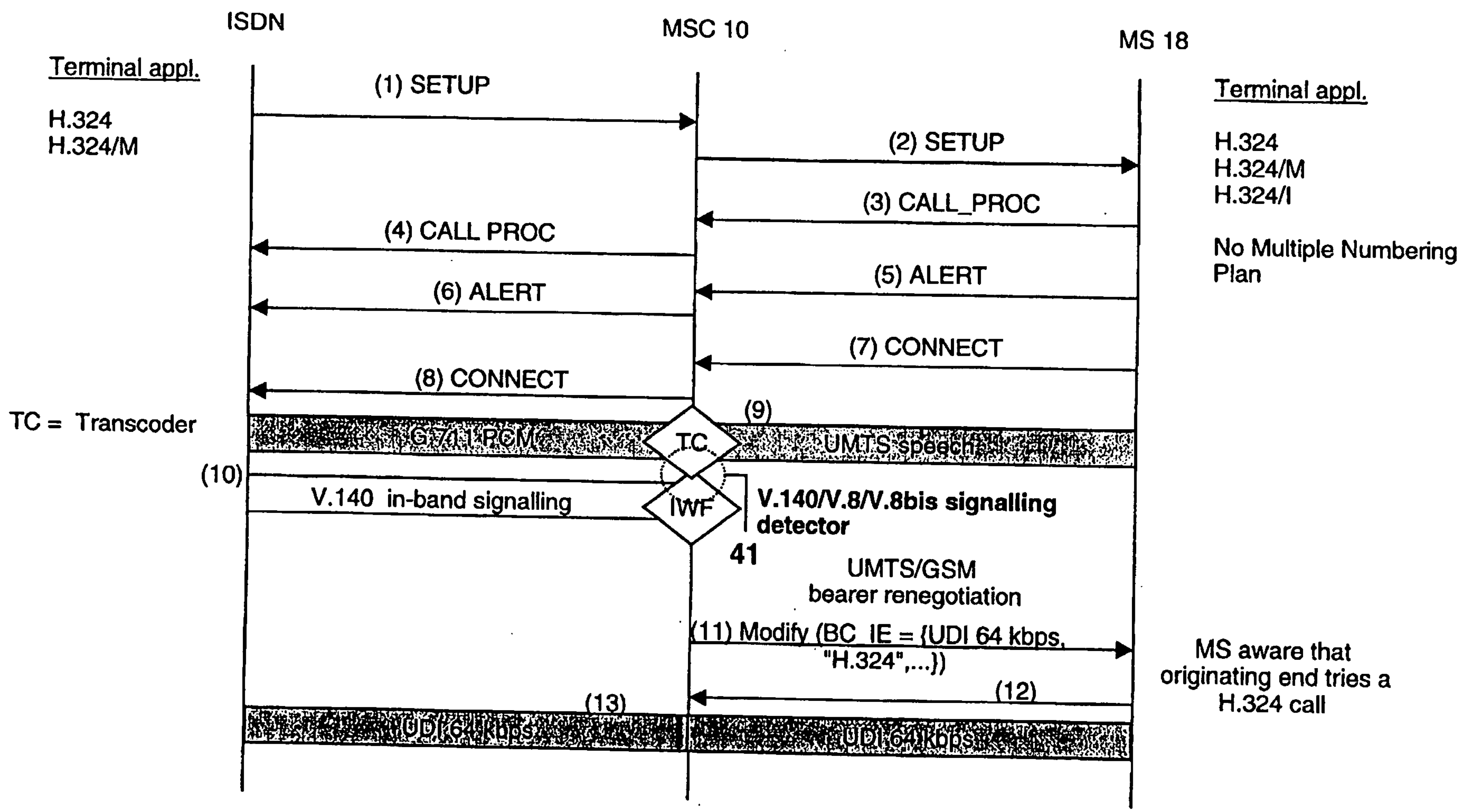


FIGURE 7b

UMTS - ISDN

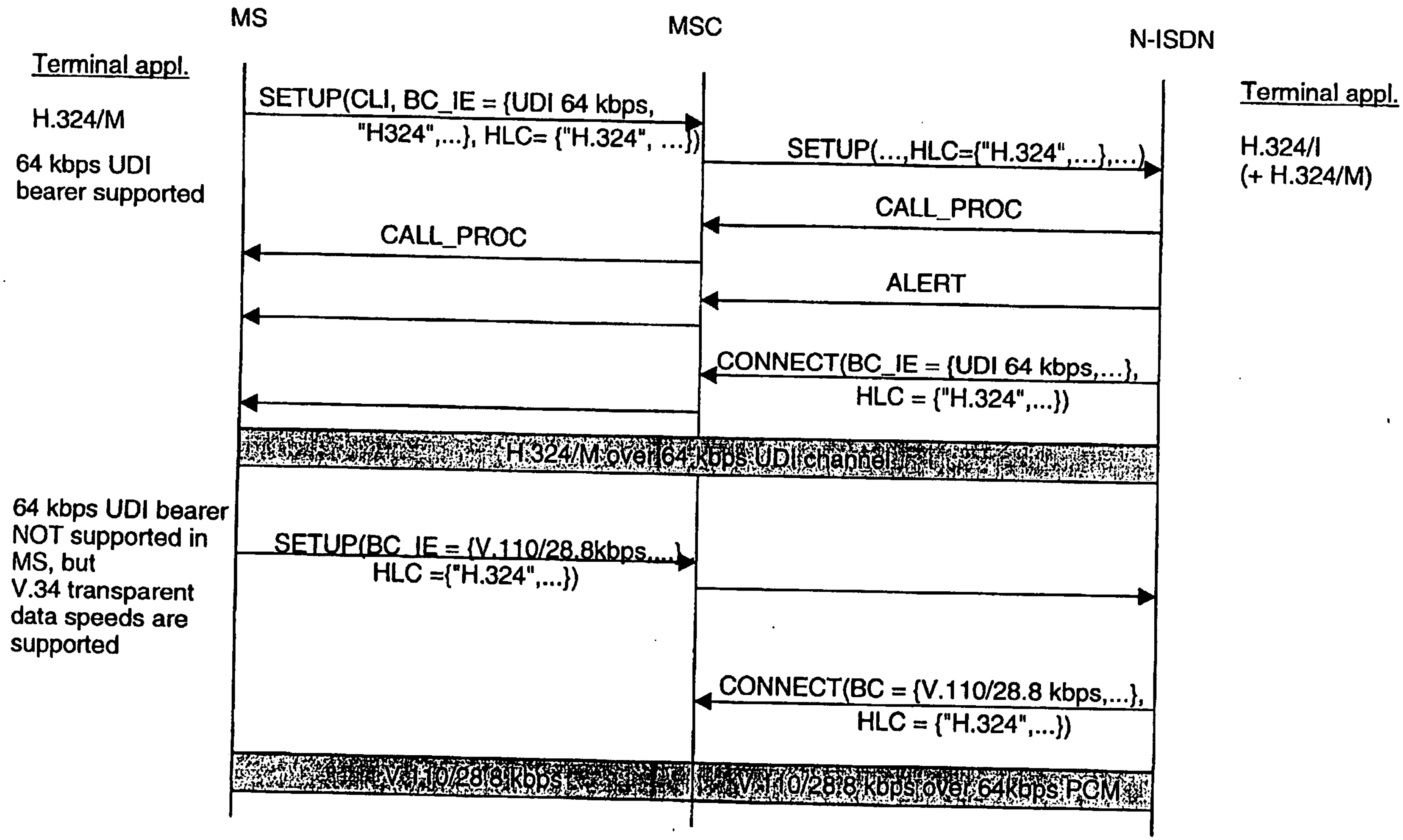


FIGURE 8a

UMTS - ISDN without end-to-end HLC

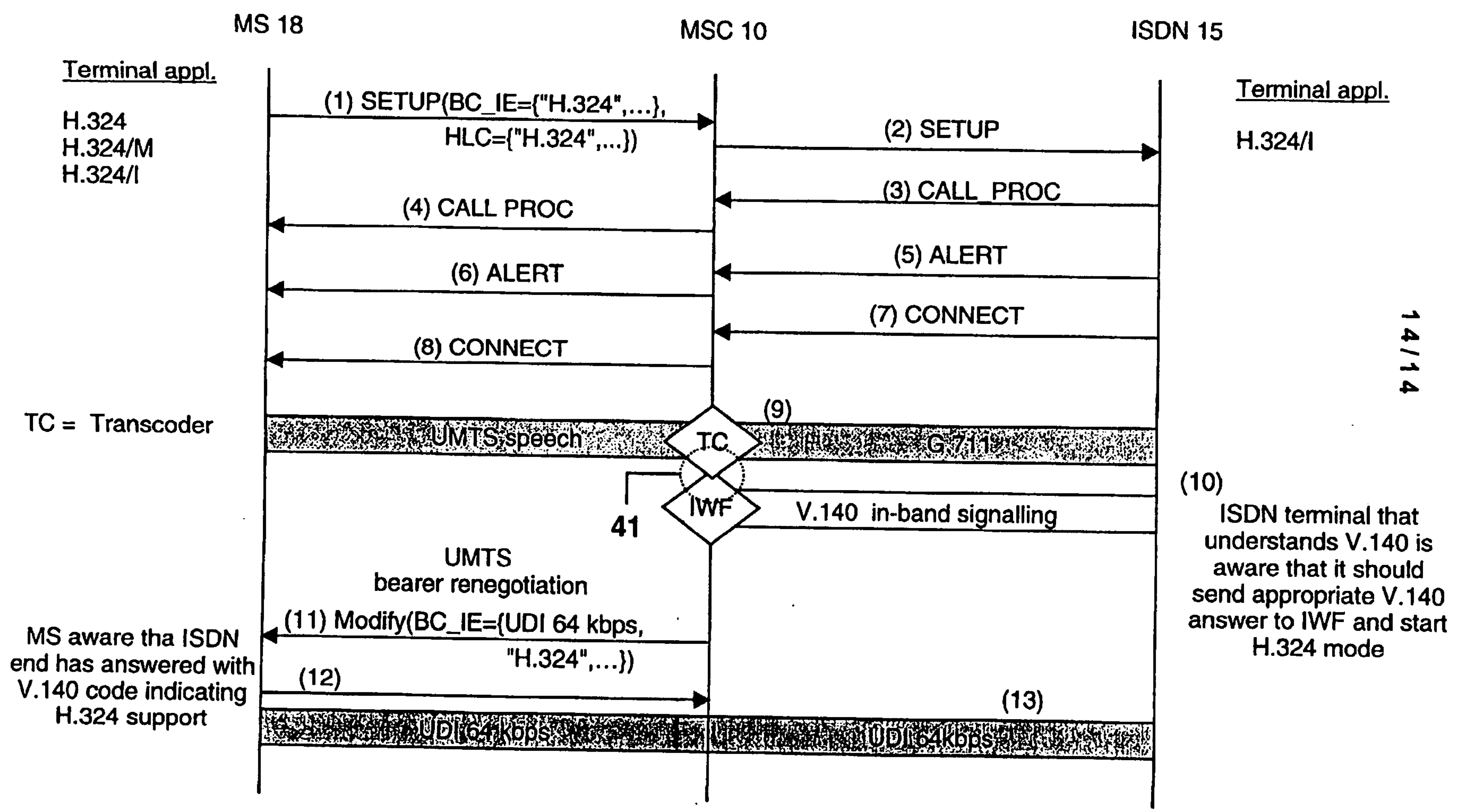


FIGURE 8b

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Telecommunication Services Identification

The present invention relates to a method and apparatus for identifying the type of telecommunications service required during call establishment over two networks. In particular, it relates to a method and apparatus for establishing call control appropriate to the telecommunications service identified by the originating network.

Traditionally, different telecommunication services have been identified by adopting a multiple numbering scheme. That is, by dedicating a specific subscriber number to a specific service. For example, in GSM, the subscriber may currently have assigned mobile terminated GSM speech, data and/or fax numbers depending upon his profile. In the future, it is proposed that he will also have the option of further high speed data and multimedia services which, too, will have dedicated subscriber numbers.

Figure 1 of the accompanying drawings illustrates the GSM public land mobile network (PLMN) and Figure 2 illustrates the call control between the public switched telephone network (PSTN) and this PLMN using the aforementioned multiple numbering scheme.

As can be seen from Figure 1, the GSM architecture comprises a gateway mobile switching centre (MSC) 10 which interfaces with fixed networks such as the PSTN 15 and a GSM radio network. The GSM radio network comprises base station systems comprising a base station controller 16 and base transceiver stations (BTS) 17. The BSS comprises a transcoder TC (e.g. for converting to GSM speech). The TC 101 is shown in the BSC in Figure 1, but could alternatively be in another part of the BSS, such as a BTS 17. Mobile stations are coupled to the BTSs via an air interface. The gateway MSC is also connected to subscriber and terminal equipment databases in the form of a home location register (HLR) 12, visitor location

register (VLR) 13, and equipment identity register (EIR) 14. The EIR contains information relating to the mobile terminals and the VLR provides a local store of all the information required to handle calls to and from mobile users in the location area relating to that particular VLR. The HLR 12 permanently stores all the user parameters of the mobile stations, including the subscriber numbers associated with a particular mobile station and their corresponding service type. Since the PSTN is an analogue network and the GSM PLMN is digital, they are not directly compatible. Hence, the gateway MSC 10 has an associated interworking function (IWF) 11, which is a functional entity enabling interworking between the PLMN and PSTN.

As illustrated in Figure 2a, when a call originates from the PSTN 15 (in this example a data call), an initial SETUP message is sent which includes the called line identification (CLI) [Step 1]. The HLR contains a database entry corresponding to this CLI indicating the call type. The MSC 10 asks the HLR 12 for this call type (bearer capability) corresponding to the CLI, and sends a setup message to the MS 18 to inform it of the incoming call and of the call type [Step 2]. The MS 18 responds by sending a CALL-PROC message after having checked the compatibility (e.g. user profile capability) with the requested bearer capability [Step 3]. This message is forwarded to the PSTN by the MSC 10 [Step 4]. Then, the MS sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the MS to the MSC [Step 7] and then from the MSC to the PSTN [Step 8] when the called subscriber answers and a connection appropriate for the data call is made.

According to one aspect of the present invention, there is provided a switch for establishing a call between a terminal of an originating network and a terminal of a terminating network, the switch comprising: an input for receiving call type information in a first format from the originating network; means for reformatting received call type information into a second format; an output

means for outputting the call type information in the second format over the terminating network; and connection means for completing a connection, suitable for the identified call type, between the terminals.
This switch enables the elimination and/or reduction of the use of the aforementioned multinumbering scheme.

The call type information may relate to teleservice and/or bearer information. The provision of bearer information enables the terminating network to use the same bitrate as is available on the originating side, thereby ensuring that bandwidth efficiency is optimised.

The terminating network is preferably a wireless communications network, such as an UMTS or GSM network.

Optionally the switch is an MSC or GMSC.

The switch may further comprise means, coupled to the input, for determining primary call type information on the basis of a subscriber number, for forwarding first primary call type information to the output, and for forwarding further primary call type information to the reformatting means. Such a switch may, for example, enable a dual numbering system to be adopted, to distinguish between calls of primary types "speech" and "data". In this case, all calls of "speech" primary type are preferably automatically connected, and further determination of actual data type is made of calls of "data" primary type.

According to another aspect of the present invention, there is provided method for establishing a call between a terminal of an originating network and a terminal of a terminating network, the method comprising: receiving call type information in a first format from the originating network; reformatting received call type information into a second format; outputting the call type information in the second format over the terminating network; and

completing a connection, suitable for the identified call type, between the terminals.

According to a further aspect of the present invention, there is provided a method for establishing a call between a terminal of an originating network and a terminal of a terminating network, the method comprising: establishing a call of a predetermined type; transmitting call type information in a first format from the originating terminal to the terminating network; reformatting received call type information into a second format; transmitting the call type information in the second format to the terminating terminal; and establishing a connection, suitable for the identified call type, between the terminals.

According to another aspect of the invention, there is provided a switching system for establishing a call from a terminal of an originating network and a terminal of a terminating network, the switching system comprising: means for receiving call type information in a first format from the originating network; means for reformatting received call type information into a second format; means for transmitting the call type information in the second format over the terminating network; and connection means for completing a connection, suitable for the identified call type, between the terminals.

According to an embodiment of the present invention an MSC is provided with a transcoder and interworking function. This is in contrast to conventional GSM systems for example, in which the transcoder forms part of the base station subsystem.

Embodiments of the present invention will now be described, by way of example, with reference to the accompanying drawings, of which:

Figure 1 illustrates the current GSM PLMN and its connection to the PSTN/ISDN;

Figure 2a illustrates data call establishment from a PSTN terminal to a mobile station using the conventional multiple numbering plan;

Figure 2b illustrates a new method of establishing a data cal from a PSTN terminal to a mobile station, still using a multiple numbering plan;

Figure 3 illustrates identification of the telecommunications service required for a PSTN originating call by an MSC of the present invention;

Figure 4a illustrates call establishment from a PSTN terminal to a mobile station using a single numbering plan according to an embodiment of the present invention;

Figure 4b illustrates call establishment from a mobile station to a PSTN terminal using V8bis signalling according to an embodiment of the present invention;

Figure 4c illustrates call establishment from a mobile station to a PSTN terminal using V8 signalling, according to an embodiment of the present invention;

Figures 5a and b illustrate. call establishment from a PSTN terminal to a mobile station using a dual numbering plan according to another embodiment of the present invention;

Figure 6 illustrates typical mapping information for an H.324 call;

Figure 7a illustrates data call establishment from an ISDN terminal to an UMTS mobile station according to an embodiment of the present invention;

Figure 7b illustrates data call establishment from an ISDN terminal to an UMTS mobile station according to another embodiment of the present invention;

Figure 8a illustrates data call establishment from an UMTS mobile station to an ISDN terminal according to an embodiment of the present invention; and

Figure 8b illustrates data call establishment from an UMTS mobile station to an ISDN terminal according to another embodiment of the present invention.

Figure 2b illustrates an example video call establishment from a PSTN terminal to a mobile station using a multiple numbering plan. The called party bearer capability information related to the called subscriber number is

fetched from the HLR. After reaching the connect phase [Step 8], the IWF modem starts listening to the CI messages and proceeds to handshake with the modem of the originating end, standard compliant PSTN H.324 application will send the V.8 CF code 0x21 [Step 9]. Only after detecting the validity of incoming V.8 code does the MSC know that originating end requests a H.324 service and is not, for example a misdialled speech call to called party number.

The invention concerns a method and apparatus for the control of different call types between incompatible networks, using unique call type differentiation. It is described with reference to the networks shown in Figure 1, namely the analogue PSTN network which uses in band signalling and the wireless digital GSM network which uses out band signalling. However, the invention is not restricted to call control over these networks, and is equally applicable to other non directly compatible networks. One area in which the invention is proposed for use in the future is between a landline network (such as the PSTN or ISDN) and universal mobile telecommunication system (UMTS).

As is illustrated in Figure 1, a conventional MSC 10 of the GSM network comprises an interworking function (IWF) 11 which interfaces with the PSTN. In this embodiment of the present invention, the MSC differs in that it has additional functionality, as outlined in Figure 3. Generally, this functionality is provided by the IWF and optionally the transducer TC. (However, alternatively, it may be provided by other parts of the MSC and/or other switches in the network).

Firstly the signalling detector 41 (comprising the TC and IWF) of this embodiment detects signalling from the PSTN [Step 301]. Then it interprets the in band signalling messages relating to the telecommunication services and maps these messages into appropriate digital out band signalling [Step 302]. Typical mapping information for an H.324 call is shown in Figure 6, and

further information on the mapping process is provided under the heading "Mapping" below.

Telecommunication services can be divided into two main types: teleservices and bearer services. Teleservices provide the full capacity for communications by means of terminal and network functions, and comprise and high layer attributes (OSI layers 1-7), whereas bearer services, in contrast, provide the capability of transmission of signals between the PSTN and PLMN access points and only involve the lower layer functions (OSI layers 1-3). The mapping of the teleservices eliminates the need for the aforementioned multinumbering scheme. It thus simplifies operability of the system from a user standpoint, as the caller only needs to remember one number for the party to be called. Also, it reduces the need to increase the length of subscriber numbers to cater for additional services subscribed to in the future. Furthermore, the processing strain on the HLR 12 is reduced. Mapping the bearer service, on the other hand, provides the additional advantage of enabling the radio bearer to be optimised. That is, the radio bearer may be negotiated to match that of the PSTN, thus optimising the bandwidth used by the wireless network. After mapping, the MSC 10 provides the appropriate call set up messaging for the identified service type [Step 303]. Consequently, the mobile station 18 is informed of the service type and the appropriate call connection is effected.

Figure 4a illustrates call establishment from a PSTN terminal to a mobile station using a single numbering plan according to a preferred embodiment of the present invention. As mentioned above, in this embodiment, the MSC 10 comprises both the IWF and TC. Also, it is adapted to conform at least with V.8, and preferably V8bis, so that it can recognise a call type from the V.8/V8bis call function information category. In particular, the MSC's modem pool includes a modem, which conforms to V.34, so as to support V8 and optionally V8bis. This modem acts as a signalling detector 41 for detecting and interpreting V8/V8bis signalling.

When a call originates from the PSTN 15 (in this example a multimedia call), an initial SETUP message is sent from the PSTN terminal to the MSC 10 [Step 1]. The MSC 10, in turn, sends a SETUP message to the MS 18 associated with the subscriber number dialled using GSM/UMTS call control signalling [Step 2]. The SETUP message informs the MS 18 of the incoming call and of a default call type: in this case speech. The MS 18 responds by sending a CALL-PROC message after having checked the compatibility with the requested bearer capability [Step 3]. This message is forwarded to the PSTN by the MSC 10 [Step 4]. Then, the MS sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the MS to the MSC [Step 7] and then from the MSC to the PSTN [Step 8] when the called subscriber answers and a default speech connection is made [Step 9] and appropriate speech transcoding is activated in the transcoder unit.

Upon connection, the V.8/V.8bis signalling detector 41 of the MSC 10 interprets the PSTN originated V.8/V.8bis signalling [Step 10]. The signalling related to the telecommunication services is mapped into corresponding GSM/UMTS signalling by the MSC 10 and, in this embodiment, the MSC instigates modification of the bearer using the GSM layer 3 call control protocol (GSM 04.08). In other words, in this case the signalling detector 41 identifies the telecommunication service categories as a multimedia teleservice category (H.324) and a 28.8kbps transparent bearer service category. In other cases more than one bearer services may be selected: for example, one for images and one for data. If necessary, connections within the MSC (including activation of modems) are rearranged for example as in the GSM Phase 1 proposed "alternate speech and data" service. The GSM/UMTS speech bearer output to the mobile station speech channel is preferably blocked to prevent loudspeaker activation in the MS until the call

type has been determined. However connection between the terminals is maintained.

Next, after successful detection of the signalling expected by the modem configuration, the modems at the MSC and in the PSTN start a handshaking process [Step 11] which results in connection with a commonly agreed data modulation rate.

Then, the bearer capability for the wireless network is set to be the same as that determined for the PSTN, so as to optimise the bandwidth, and the MS 18 is informed of the requested call type [Step 12]. The MS returns a message to the MSC to complete the GSM/UMTS bearer modification process [Step 13]. The connection is now complete for multimedia communication between the PSTN terminal and MS 18 [Step 14]. Communication takes place in 3.1 kHz (external to the PLMN) bearer service using the category involving the IWF modem and optimised radio bearer [Step 14].

Figure 4b illustrates call establishment from an UMTS mobile station to a PSTN terminal (Mobile originated call setup) using a single numbering plan according to an embodiment of the present invention. Again, in this embodiment, the MSC 10 is adapted to conform with V8bis, so that it can send and recognise a call type from the V8bis call function information category. In particular, for PSTN interworking the MSC's modem pool includes a modem, which conforms to V.34, so as to support V8bis. This modem-transcoder unit of MSC acts as a signalling source and detector 41 for detecting and interpreting V8bis signalling.

When a call originates from the MS 18 (in this example a multimedia call), an initial SETUP message is sent from the MS 18 to the MSC10. This message includes the video service specific codepoint by default in both BC and HLC /LLC information elements [Step 1]. The MSC 10 interprets the requested

service from the BC information element and sends a SETUP message to the PSTN terminal 15 [Step 2]. In the situation where the called party resides within digital call control domain (ISDN, other PLMN), the terminal interprets the requested service from the peer-to-peer transmitted HLC/LLC information element. However in the current case, the SETUP message [Step 2] does not forward digital call control specific information to PSTN terminal 15 to indicate the requested bearer service. Instead, it merely indicates a default call type using the BC element. The PSTN terminal responds by sending a CALL-PROC message, without indication of the expected bearer service [Step 3]. Thus, a connection is completed with knowledge by the MSC 10 that called party resides behind PSTN network [Steps 4 – 8]. That is, the CALL-PROC message is forwarded to the mobile station 18 by the MSC 10 [Step 4]. Then the PSTN terminal sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the PSTN terminal to the MSC [Step 7] and then from the MSC to the MS 18 [Step 8] when the called subscriber answers and a default speech connection is made [Step 9]. In other words, as both H.324 speech call first and direct video service signalling could be anticipated from the MS, the default user data (here default is speech service) is switched to pass through a transcoder-voice modem unit of MSC10. At the time the mobile station wishes to start the transition to video call (immediately or later at the wish of mobile station user), the DCE in signalling detector 41 is told by the PLMN to send appropriate V.8bis signalling (for example as described in Intel's Videophone Ready Modem Handbook Revision 1.1 & the H.324 specification) towards the PSTN terminal 15 [Step 10]. When the appropriate signalling is interpreted in the PSTN terminal 15, it switches to the appropriate DTE-DCE mode and starts modem handshaking with involved voice modem in MSC 10 [Step 11] This results in a data connection between modems with a common data modulation.

Next, the bearer capability for the wireless network is set to be the same as that determined for the PSTN, so as to optimise the bandwidth, and the MS 18 is informed of the requested call type [Step 12]. The MS also gets a confirmation that it has received a acknowledge for its original video call mode change request, and returns a message to the MSC to complete the UMTS bearer modification process [Step 13]. The connection is now complete for multimedia communication between the MS 18 and PSTN terminal [Step 14]. Communication takes place in 3,1 kHz (external to the PLMN) bearer service category [Ref: GSM 2.02] involving modem at the interworking function of MSC 10 and using the optimised radio bearer.

Figure 4c illustrates call establishment from an UMTS mobile station 18 to a PSTN terminal according to an embodiment of the present invention. More particularly, it shows establishment of an MS 18 originated video call with direct video initialisation with support of (BC information element information based) MSC – configured V8 signalling which originates in a data mode enabled modem of the MSC 10. In this embodiment, the MSC 10 comprises the IWF. Also, it is adapted to conform with V.8, (but not V8bis), so that it can recognise a call type from the V.8 call function information category. In particular, the MSC's modem pool includes a modem, which conforms to V.34, so as to support V8. This modem acts as a signalling detector 41 for detecting and interpreting V8 signalling.

When a call originates from the MS 18 (in this example a multimedia call), an initial SETUP message is sent from the MS 18 to the MSC 10 [Step 1]. The MSC 10, in turn, sends a SETUP message to the PSTN terminal 15 associated with the subscriber number dialled. [Step 2]. The SETUP message informs the PSTN terminal 15 of the incoming call and of a default call type: in this case speech. The PSTN terminal 15 responds by sending a CALL-PROC message after having checked the compatibility with the requested bearer capability [Step 3]. This message is forwarded to the mobile station 18 by the MSC 10 [Step 4]. Then, the PSTN terminal sends

and ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Next, a CONNECT message is sent from the PSTN terminal 15 to the MSC [Step 7].

The IWF of the MSC 10 receives the signalling [Step 8]. In the event that it receives V.8 CL code 00x21, this is interpreted by the IWF and mapped into corresponding UMTS signalling by the MSC 10. Then, the bearer capability for the wireless network is set to be the same as that determined for the PSTN, and the connection is effected for multimedia communication between the MS 18 and PSTN terminal [Step 9a]. Communication takes place via the MSC modem pool using the optimised radio bearer [Step 10a].

Alternatively, in the event that the IWF of the MSC 10 does not receive V8 CL code 00x21 (or indeed any mobile signalling), the MSC 10 identifies the telecommunication service as a speech service and the MSC 10 maps the associated signalling into UMTS signalling. Then a connection message is sent for speech to the MS 18 [Step 9b] and a speech connection is established between the MS 18 and PSTN terminal 15 [Step 10b].

An alternative embodiment of the invention enabling the use of a reduced numbering plan will now be described. Figures 5a and 5b illustrate call establishment from a PSTN terminal to a mobile station using a dual numbering plan. This embodiment provides a hybrid arrangement in which only two numbers are used: one to identify speech calls and the other to identify data calls. As in the arrangement of Figure 2, the HLR 12 contains a database entry corresponding to the called line identification indicating the call type. However, in this embodiment, this database has only two entries: one indicating that the call type is speech and the other indicating that the call type is data. If the call type is speech, the PSTN originating signal is transcoded by the TC 101 of the MSC 10 and transmitted over the radio

network. However, if it is data, the actual type is determined by the IWF 11 as described with reference to Figure 3 above.

Call establishment in this embodiment will now be described in more detail, with reference to Figures 5a and b.

Figure 5a illustrates call establishment for a speech call. When a call originates from the PSTN 15, an initial SETUP message is sent which includes the called line identification (CLI) – in this case for a speech call. [Step 1]. As mentioned above, the HLR contains a database entry corresponding to this CLI indicating the call type as speech. The MSC 10 asks the HLR 12 for this call type (bearer capability) corresponding to the CLI, and sends a setup message to the MS 18 to inform it of the incoming call and that the call type is speech [Step 2]. The MS 18 responds by sending a CALL-PROC message after having checked the compatibility (e.g. user profile capability) with the requested bearer capability [Step 3]. This message is forwarded to the PSTN by the MSC 10 [Step 4]. Then, the MS sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the MS to the MSC [Step 7] and then from the MSC to the PSTN [Step 8] when the called subscriber answers and a connection appropriate for the speech call is made.

Figure 5b illustrates call establishment for a data call. When a call originates from the PSTN 15 (in this example a multimedia call), an initial SETUP message is sent which includes the called line identification (CLI) – in this case for a data call. [Step 1]. As mentioned above, the HLR contains a database entry corresponding to this CLI indicating the call type as data. The MSC 10 asks the HLR 12 for this call type (bearer capability) corresponding to the CLI, and sends a setup message to the MS 18 to inform it of the incoming call and of a default call type (e.g. a default data type) [Step 2]. The MS 18

responds by sending a CALL-PROC message after having checked the compatibility with the requested bearer capability [Step 3]. This message is forwarded to the PSTN by the MSC 10 [Step 4]. Then, the MS sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the MS to the MSC [Step 7] and then from the MSC to the PSTN [Step 8] when the called subscriber answers and the default connection is made [Step 9].

Next, the MSC endeavours to determine the actual data type. In this example, upon connection, the V.8/V.8bis signalling detector 41 of the MSC 10 interprets the PSTN originated V.8/V.8bis signalling [Step 10]. The signalling related to the telecommunication services is mapped into corresponding GSM/UMTS signalling by the MSC 10 and, in this embodiment, the MSC instigates modification of the bearer using the GSM layer 3 call control protocol (GSM 04.08). In other words, in this case the detector 41 identifies the telecommunication service categories as a multimedia teleservice category (H.324) and a 28.8kbps transparent bearer service category. If necessary, connections within the MSC (including activation of modems) are rearranged for example as in the GSM Phase 1 proposed "alternate speech and data" service. The GSM/UMTS speech bearer output to the mobile station speech channel is preferably blocked to prevent loudspeaker activation in the MS until the call type has been determined. However connection between the terminals is maintained.

Next, modem handshaking takes place [Step 11]. The, the bearer capability for the wireless network is set to be the same as that determined for the PSTN, so as to optimise the bandwidth, and the MS 18 is informed of the requested call type [Step 12]. The MS returns a message to the MSC to complete the GSM/UMTS bearer modification process [Step 13]. The connection is now complete for multimedia communication between the PSTN

terminal and MS 18 [Step 14]. Communication takes place via the MSC modem pool using the optimised radio bearer.

Figures 7 and 8 illustrate calls between an ISDN terminal and UMTS mobile station, figure 7a and b from an ISDN terminal 15 to a MS 18 and figures 8a and b vice versa.

Figure 7a illustrates a digital domain call setup with peer-to-peer HLC assuming global acceptance of HLC value for H.324 service.

Figure 7b illustrates call establishment from a ISDN terminal 15 to a mobile station 18 using a single numbering plan according to another embodiment of the present invention. Usually in digital network domain the call control is based on peer-to-peer signalling with indication of e.g. the preferred call capability in HLC information element. Networks exist which fail to send out-band signalling messages end-to-end, and thus in this embodiment, the transcoder in MSC 10 is adapted to conform with V.140 (an ISDN in-band signalling protocol comparable to V.8/V.8bis in PSTN), so that it can recognise a call type from the in-band transmitted V.140 information. In particular, the MSC's modem-transcoder 41, which can detect V.140 signalling in LSBs of 56/64 kbps PCM modulated data as explained in the V.140 specification. This modem-transcoder unit of MSC acts as a signalling detector 41 for detecting and interpreting V.140 signalling.

When a call originates from the ISDN terminal 15 (in this example a multimedia call), an initial SETUP message is sent from the ISDN terminal to the MSC 10 [Step 1]. The MSC 10, in turn, sends a SETUP message to the MS 18 associated with the subscriber number dialled using GSM/UMTS call control signalling [Step 2]. The SETUP message informs the MS 18 of the incoming call and of a default call type: in this case speech with 3,1 kHz bearer service category (GSM 02.02). The MS 18 responds by sending a CALL-PROC message after having checked the compatibility with the

requested bearer capability [Step 3]. This message is forwarded to the ISDN by the MSC 10 [Step 4]. Then, the MS sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the MS to the MSC [Step 7] and then from the MSC to the ISDN [Step 8]. When the called subscriber answers and a default speech connection is made [Step 9], appropriate speech transcoding is activated in the transcoder unit.

Upon connection, the V.140 signalling detector 41 of the MSC 10 interprets the ISDN originated V.140 signalling [Step 10]. The detected signalling related to the telecommunication services is mapped into corresponding GSM/UMTS signalling by the MSC 10 and, in this embodiment, the MSC instigates modification of the bearer using the layer 3 call control protocol. In other words, the detector 41 identifies the telecommunication service categories as a multimedia teleservice category (H.324) and a 64kbps transparent UDI bearer service category.

Then, the bearer capability for the wireless network is set to be the same as that determined for the ISDN, and the MS 18 is informed of the requested call type in the BC information element [Step 11]. The MS returns a message to the MSC to complete the GSM/UMTS bearer modification process [Step 12]. The connection is now complete for UDI multimedia communication between the PSTN terminal and MS 18 [Step 13].

Figure 8a illustrates a digital domain setup with peer-to-peer HLC.

Figure 8b illustrates call establishment from a mobile station to an ISDN terminal (Mobile originated call setup) according to an embodiment of the present invention. In this embodiment, the transcoder in MSC 10 is adapted to conform with V.140 so that it can indicate a call type within the in-band transmitted V.140 information. In particular, the MSC's modem-transcoder

41, which can indicate V.140 signalling in LSBs of 56/64 kbps PCM modulated data as explained in the V.140 specification. This modem-transcoder unit of MSC acts as a signalling source 41 for transmitting, detecting and interpreting V.140 signalling.

When a call originates from the MS 18, an initial SETUP message is sent which includes the video service specific codepoint by default in both BC and HLC/LLC information elements [Step 1]. The MSC 10 interprets the requested service from the BC information element and in optimal case the called party would reside within digital call control domain (ISDN, other PLMN) end terminal would interpret it from the peer-to-peer transmitted HLC/LLC information element. However in a network environment which does not confirm to ISUP, the SETUP message [Step 2] does not forward the digital call control specific information to the ISDN terminal. The ISDN terminal responds by sending a CALL-PROC message [Step 3].

Thus, connection signalling is completed with knowledge by MSC 10 that called party resides behind either ISDN or PSTN network [Steps 4 – 8]. As both H.324 speech call first and direct video service signalling could be anticipated from the MS 18, the default user data (here default is speech service) is switched to pass through a transcoder-voice modem unit 41 of MSC10 [Step 9]. At the time the mobile station wishes to start transition to video call (immediately or later at the wish of mobile station user), the transcoder in signalling detector 41 shall be told by the PLMN to send appropriate V.140 (and/or simultaneous V.8bis signalling) as described in V.140] (& Intel's Videophone Ready Modem Handbook & H.324 specification) towards the ISDN terminal 15 [Step 10]. When the appropriate signalling is interpreted in ISDN terminal, it will switch to appropriate UDI mode and activate the video application in its own end.

Next, the bearer capability for the wireless network is set to be the same as that determined for the PSTN, so as to optimise the bandwidth, and the MS

18 is informed of the requested call type within the BC information element [Step 11]. The MS also gets a confirmation that it has received a acknowledge for its original video call mode change request. The MS 18 returns a message to the MSC to complete the GSM/UMTS bearer modification process [Step 12]. The connection is now complete for multimedia communication between the PSTN terminal and MS 18 [Step 13]. Communication takes place in 64 kbps UDI bearer service category.

Mapping

Typical mapping by the MSC will now be described, firstly from V.8 to GSM and then optional mapping for V.8bis. As will be appreciated, this mapping is only exemplary and similar mapping could take place for future networks such as UMTS. GSM signalling is generally derived from ISDN, i.e. it is based on ITU-T Q.931 recommendation. So too are 3G telecommunication systems such as UMTS, hence, their particular compatibility with such mapping methods.

In this embodiment, information relating to the teleservice is mapped into the HLC (high level capability) information element in GSM (based on Q.931). Similarly, information relating to the bearer service is mapped into the BC (bearer capability) information element in GSM. Further details of the HLC, BC and also the LLC (low level capability element) are also provided in the following description of the mapping.

The V.8, V.8bis and GSM signalling information, which is relevant to the determination of telecommunication service types, is outlined below. The references refer to the following documents, the contents of which are incorporated herein by reference:

[1] ITU-T Recommendation F.721 (08/92) - Videotelephony teleservice for ISDN

[2] ITU-T Recommendation F.700 (07/96) - Framework Recommendation for audiovisual/multimedia services

[3] ITU-T Recommendation V.8 (02/98) - Procedures for starting sessions of data transmission over the public switched telephone network

[4] ITU-T Recommendation V.8bis (08/96) - Procedures for the identification and selection of common modes of operation between data circuit-terminating equipment (DCEs) and between data terminal equipment (DTEs) over the general switched telephone network and on leased point-to-point telephone-type circuits

[5] ITU-T Recommendation H.324 (2/98) - Terminal for low bit rate multimedia communication

[6] Digital cellular telecommunications system (Phase 2+); Mobile radio interface layer 3 specification, GSM 04.08 version 6.2.0 Release 1997

[7] ITU-T Recommendation H.245 (2/98) - Control protocol for multimedia communication.

V.8 signalling to be mapped [3]

Table 2 in [3] lists the information categories that are available in V.8 signalling. As can be seen, the call-function octet has 3 option bits, and Table 3 in [3] illustrates how these option bits are used to identify particular call functions. As mentioned above, the information interpreted from the call function category can be mapped into the Q.931 HLC information element to

provide the GSM network with information pertaining to the teleservice or call function and the bearer service.

Table 4 in [3] indicates the availability of PSTN V-Series modulation modes other than V.90. The availability is only shown if the modulation mode can be used with the indicated call function, and if it is desired to convey that capability to the remote DCE. For example, if an H.324 application is used, the use of a V.34 full duplex mode is mandatory, and the other options would not be indicated. The modulation mode information is only used between DCEs in order to find out the common modulation modes. After selection of a common modulation mode (e.g. V.34), the MSC/IWF determines the data rate from the modem (e.g. 28.8kbps) on the basis of how good the quality (BER) of the PSTN line is.

GSM (Q931 based) signalling onto which the above V8 signalling is mapped by the MSC

GSM specification 04-08 [6] defines the messages for circuit switched call control. In particular, section 9.3.23.1 in [6] defines the SET UP message content for mobile terminated call establishment. This message is sent over the network to the mobile station to initiate mobile terminated call establishment. The SET UP message contains relevant information elements for the BC (bearer capability), HLC (high level capability) and LLC (low level capability). As mentioned above, it is these information elements which the V8 call function category information is mapped into. (In GSM at least, not all of these information elements are compulsory. However, in UMTS, at least the BC and HLC should be.)

BC

The purpose of the bearer capability information element is to describe the bearer service to be used in the connection. Figure 10.5.88 in [6] illustrates

the BC element for GSM. This element contains 16 octets (8 bit units). Certain bits or coding points need to be stored in octets 3, 4, 6, 6a, 6c and 6d in order to support H.324, as outlined below.

Transfer mode (octet 3):

Bit

4

0 circuit mode

1 packet mode

- Circuit mode is selected

Duplex mode (octet 4)

Bit

4

0 half duplex

1 full duplex

- Full duplex is selected

Synchronous/asynchronous (octet 6)

Bit

1

0 synchronous

1 asynchronous

- Synchronous is selected

User rate (octet 6a)

Bits

4 3 2 1

0 0 0 1 0.3 kbit/s Recommendation X.1 and V.110

0 0 1 0 1.2 kbit/s Recommendation X.1 and V.110

0 0 1 1 2.4 kbit/s Recommendation X.1 and V.110

0 1 0 0 4.8 kbit/s Recommendation X.1 and V.110

0 1 0 1 9.6 kbit/s Recommendation X.1 and V.110

0 1 1 0 12.0 kbit/s transparent (non compliance with X.1 and V.110)

0 1 1 1 1.2 kbit/s/75 bit/s Recommendation V.23, (asymmetric) X.1, V.110.

The coding points shown above are used to indicate the user rate. At present there is no e.g. 28.8 kbps user rate available in GSM. However, Q.931 offers a wider set of user data rates that can be used when specifying the BC information element. (An outline of how the mapping might occur for the 28.8kbps and other unspecified data rates in GSM is outlined under the heading "Potential BC for other user rates".

Connection element (octet 6c)

Bit

7 6

0 0 transparent

0 1 non transparent (RLP)

1 0 both, transparent preferred

1 1 both, non transparent preferred

- Transparent is selected

Other modem type (octet 6d)

Bits

7 6

0 0 no other modem type specified in this field

0 1	V.32bis
1 0	**V.34**

- V.34 is selected

<u>Potential BC for other bit user rates.</u>

Fixed network user rate (octet 6d)

Bit

5 4 3 2 1	
0 0 0 0 0	Fixed network user rate not applicable/No meaning is associated with this value.
0 0 0 0 1	9.6 kbit/s Recommendation X.1 and V.110
0 0 0 1 0	14.4 kbit/s Recommendation X.1 and V.110
0 0 0 1 1	19.2 kbit/s Recommendation X.1 and V.110
0 0 1 0 0	**28.8 kbit/s** Recommendation X.1 and V.110
0 0 1 0 1	38.4 kbit/s Recommendation X.1 and V.110
0 0 1 1 0	48.0 kbit/s Recommendation X.1 and V.110(synch)
0 0 1 1 1	56.0 kbit/s Recommendation X.1 and V.110(synch) /bit transparent
0 1 0 0 0	64.0 kbit/s bit transparent

For an example data rate of 28.8 kbps, the element highlighted above might be used. So 28.8 is selected.

Acceptable channel codings (octet 6e), mobile station to network direction:

Bit

7	
0	TCH/F14.4 not acceptable
1	**TCH/F14.4 acceptable**

If 14.4 is implemented then it can be selected.

Bit

6

0 Spare

Bit

5

0 TCH/F9.6 not acceptable

1 **TCH/F9.6 acceptable**

Normally, 9.6 is implemented as well

Bit

4

0 TCH/F4.8 not acceptable

1 **TCH/F4.8 acceptable**

4.8 exists even though not widely used

Acceptable channel codings (octet 6e), network to MS direction:

Bits 4 to 7 are spare and shall be set to "0".

This would mean that network can not decide what kind of channel coding is used. This is required in the present case since networks initiate the call setup renegotiation. A change to current GSM implementation is needed to effect this.

Maximum number of traffic channels (octet 6e), MS to network direction:

Bits

3 2 1

0 0 0 1 TCH

0 0 1 2 TCH

0 1 0 3 TCH

0 1 1 4 TCH

1 0 0 5 TCH

1 0 1 6 TCH

1 1 0 7 TCH

1 1 1 8 TCH

28.8kbps can be obtained by combining two 14.4 channels using HSCSD. This could also be done by combining three 9.6 channels.

Maximum number of traffic channels (octet 6e), network to MS direction:
Bits 1 to 3 are spare and shall be set to "0".
As above, this would mean that network is unable to decide what kind of channel coding is used, and would require a modification to GSM.

BC Information derived from Q931

BC information used in GSM inband signalling is derived from Q.931 and modified to GSM purpose. The BC element discussed above is proposed to be used in UMTS in a frame of reference of this patent application.

In Q.931 BC information element there exists one additional coding point which is needed and should be included to UMTS signalling. This coding point is User information layer 1 protocol (octet 5) and is described below:

User information layer 1 protocol (octet 5)
Bits

54 321	
00001	CCITT standardised rate adaption V.110, I.460 and X.30. This implies the presence of octet 5a and optionally octets 5b, 5c and 5d as defined below.
00010	Recommendation G.711 μ-law
00011	Recommendation G.711 A-law
00100	Recommendation G.721 32 kbit/s ADPCM and Recommendation I.460.
00101	Recommendations H.221 and H.242

00110	**Recommendations H.223 and H.245**
00111	Non-ITU-T standardized rate adaption. This implies the presence of octet 5a and, optionally, octets 5b, 5c and 5d. The use of this code point indicates that the user rate specified in octet 5a is defined by the user. Additionally, octets 5b, 5c and 5d, if present, are defined consistent with the user specified rate adaption.
01000	ITU-T standardised rate adaption V.120. This implies the presence of octets 5a and 5b as defined below, and optionally octets 5c and 5d.
01001	CCITT standardised rate adaption X.31 HDLC flag stuffing.
	All other values are reserved.

- Recommendations H.223 and H.245 is selected

LLC

The low layer compatibility information element is illustrated in Figure10.5.104 of [6]. The purpose of the low layer compatibility information element is to provide a means which should be used for compatibility checking by an addressed entity (e.g., a remote user or an interworking unit or a high layer function network node addressed by the calling user). The low layer compatibility information element is transferred transparently by a PLMN between the call originating entity (e.g. the calling user) and the addressed entity.

Except for the information element identifier, the low layer compatibility information element is coded as in ETS 300 102-1.

This information element is not necessarily needed because the same coding can be conveyed in the BC.

HLC

The purpose of the high layer capability information element is to provide a means by which the remote user can check for compatibility.

The high layer compatibility information element is coded as shown in figure 10.5.102 and table 10.5.125 of [6].

The high layer compatibility information element is transported transparently by a PLMN between a call originating entity (e.g. a calling user) and the addressed entity (e.g. a remote user or a high layer function network node addressed by the call originating entity). However, if explicitly requested by the user (at subscription time), a network which provides some capabilities to realise teleservices may interpret this information to provide a particular service.

The following HLC code points can be found from the Q.931 recommendation:

High layer characteristics identification (octet 4)
Bits
7 6 5 4 3 2 1
0 0 0 0 0 0 1 Telephony
0 0 0 0 1 0 0 Facsimile Group 2/3 (Recommendation F.182 [68])
0 1 0 0 0 0 1 Facsimile Group 4 Class I (Recommendation F.184 [69])
0 1 0 0 1 0 0 Teletex service, basic and mixed mode of operation (Recommendation F.230 [70]) and facsimile service Group 4, Classes II and III (Recommendation F.184)
0 1 0 1 0 0 0 Teletex service, basic and processable mode of operation (Recommendation F.220 [71])
0 1 1 0 0 0 1 Teletex service, basic mode of operation (Recommendation F.200 [72])

0 1 1 0 0 1 0 Syntax based Videotex (Recommendations F.300 [73] and T.102 [74])

0 1 1 0 0 1 1 International Videotex interworking via gateways or interworking units (Recommendations F.300 and T.101 [75])

0 1 1 0 1 0 1 Telex service (Recommendation F.60 [76])

0 1 1 1 0 0 0 Message Handling Systems (MHS) (X.400 - Series Recommendations [77])

1 0 0 0 0 0 1 OSI application (Note 2) (X.200 - Series Recommendations [78])

1 0 1 1 1 1 0 Reserved for maintenance (Note 4)

1 0 1 1 1 1 1 Reserved for management (Note 4)

1 1 0 0 0 0 0 Audio visual (Recommendation F.721 [79])

1 1 0 0 0 0 1 through 1 1 0 1 1 1 1 Reserved for audiovisual services [2]

1 1 1 1 1 1 1 Reserved

The F.700 Recommendation [2] provides a methodology for constructing multimedia services which is timely and responsive to the needs of both the End-User and Service Provider. This methodology links generic End-User application requirements with the established set of network independent building blocks or other communications capabilities being developed within ITU-T. The procedures described in this Recommendation are intended for use in developing a series of General Service Recommendations which combine the required communication tasks and media components into an architecture for generic services (e.g. for Multimedia Conferencing Service, Multimedia Distribution Service, etc.). Where applicable Recommendations are not yet available, this methodology can be used as the basis for a structured dialogue between End-Users and Service Providers in arriving at a responsive service solution.

Bits 1100001 through 1101111, specified in F.700, could be used for e.g. H.324 code point or a totally new codepoint could be designed for UMTS use.

<u>V.8bis signalling to be mapped [4]</u>

Table 6-2 in [4] lists the standard information categories that are available in V.8bis signalling. It comprises categories such as data and H.324 multimedia terminal to readily cater for such future teleservices. The coding for these teleservices are illustrated in Tables 6-3 and 6-5 of [4]. One advantage of the V.8bis capability exchange is that, in most cases, it enables terminals to ascertain very quickly when operation in H.324 mode is desired. This is because V.8bis capabilities indicate only the most basic and commonly used modes. If an H.324 operation mode not signalled by V.8bis is desired, the terminal must complete call establishment and perform a H.245 [7] capabilities exchange to determine if the far-end terminal supports the desired mode.

Within the Rec. V.8bis Communications Capabilities (CC) field for H.324, the CC field is formatted into one or more sub-fields. Each sub-field ends with the octet in which bit [n] is set to 1. Following the first sub-field, the remaining sub-fields, if present, shall appear in the same order in which the bits indicating their presence are transmitted. The actual bit assignments can be seen from [4].

In the first sub-field the following bits are allocated:

Name	Meaning
Video	Shall be set only if bi-directional video is supported per Rec. H.324 (sec. 6.6).
Audio	Shall be set only if bi-directional audio is supported per Rec. H.324 (sec. 6.7).
Encryption	Shall be set only if encryption is supported per Rec. H.324 (sec. 9.2).
Data	Indicates that a data subfield is present. Shall be set only if one or more bits in the data subfield are set.

Possible future allocations include Profiles (new subfield).

In the Data subfield, the following bits are allocated:

Name	Meaning
T.120	Shall be set only if T.120 conferencing is supported per Rec. H.324 (sec. 6.8.2.1).
T.84	Shall be set only if T.84 still image transfer is supported per Rec. H.324 (sec. 6.8.2.2).
T.434	Shall be set only if T.434 file transfer is supported per Rec. H.324 (sec. 6.8.2.3).
V.42	Shall be set only if V.42 user data is supported per Rec. H.324 (sec. 6.8.1.2/6.8.2.6).
V.14	Shall be set only if V.14 user data is supported per Rec. H.324 (sec. 6.8.1.1/6.8.2.6).
PPP	Shall be set only if IETF Point-to-Point protocol is supported via the Network Layer Protocol Identifier (NLPID) per Rec. H.324 (sec. 6.8.2.5).
T.140	Shall be set only if T.140 Text Conversation Protocol for Multimedia Application is supported per Rec H.324 (sec 6.8.2.8).

Other modes beside those indicated in V.8bis, such as unidirectional modes, may be supported by terminals as signalled via H.245 capabilities exchange.

With V.8bis signalling can also be used when the call is first started in speech mode and after that it is switched to e.g. H.324 mode. For this reason, the MSC/IFW must all the time be able to listen the possible inband signalling coming from PSTN modem. The switch of service then initiates the bearer renegotiation in UMTS side where the "old" bearer is accomplished according to QoS parameters needed to H.324 call.

Mapping between UMTS and PSTN/ISDN

In this section different information elements and their codings are described. These codings must be mapped in the interface of either PSTN/UMTS or ISDN/UMTS.

Necessary PSTN codings in the frame of reference of the patent application:

V.8

	Call function category (Octet – "callf0")
100	PSTN Multimedia Terminal (Bits 567)
	Modulation modes category (Octet – "modn0")
1	V.34 duplex availability (Bit 6)

V.25ter

	Modulation control commands
	Modulation reporting control (+MR)
+MCR: V34	+MCR: <carrier>
+MRR: 28800	+MRR: <rate>

V.8bis

	Standard information field - {SPar(1)} coding
	Standard information field – Data {NPar(2)} coding (Octet 2)
1	Rec. V.34 duplex mode (Bit 5)
	Standard information field - {SPar(1)} coding
	Standard information field – H.324 multimedia terminal {NPar(2)} coding
1	Video (Bit 1)
	Standard information field - {SPar(1)} coding
	Standard information field – H.324 multimedia terminal {NPar(2)} coding
1	Audio (Bit 2)

Necessary ISDN codings (in a case V.140 signalling is needed) in the frame of reference of the patent application:

V.140 inband signals consist of HDLC-framed PDUs defined using ASN.1 syntax according to Recommendation X.680 and coded according to the packed encoding rules of Recommendation X.691.

The purpose why V.140 signalling is used is that there exist networks that fail to send out band signalling messages end to end. V.140 signalling used in ISDN does more or less the same thing as V.8/V.8bis signalling in PSTN.

The capabilitySet field of the roleAndCapability PDU contains a sequence of one or more Capability structures, each expressing the terminal's ability to work in a particular multimedia or other communication protocol. The transmitter shall include the complete list of modes in which it is currently able to operate. The list of possible modes is defined in Annex A of V.140, and may be extended in the future. Capabilities shall be listed in order of preference, from most-preferred to least-preferred. In our case the h324AnnexD should be the first (or only one) of the listed capabilities in capabilitySet field.

The receiving terminal can answer the transmitting terminal of the selected mode by using modeSelect PDU.

In the following the syntax of PDUs using ASN.1 notation is described [V.140 Annex A]. The "code point" of our choice is presented with bigger font.

```
HDISPATCH DEFINITIONS AUTOMATIC TAGS ::=
BEGIN

-- Export all symbols

--
==========================================================
-- Top level PDUs
--
==========================================================

HDispatchPDU                    ::=CHOICE
{
      nonStandard               NonStandardMessage,
      roleAndCapability         RoleAndCapabilityMessage,
      modeSelect                Mode,
      youChoose                 NULL,
      modeSelectAcknowledge     NULL,
      terminate                 TerminateMessage,
      ...
}

RoleAndCapabilityMessage        ::=SEQUENCE
{
```

```
    role                        CHOICE
    {
        originate               NULL,
        unknown                 NULL,
        answer                  NULL,
        ...
    },

    arbitrationField            INTEGER (0..4294967295), -- 32 bit
                                random #
    capabilitySet               SEQUENCE SIZE (1..65535) OF
                                Capability,
    ...
}

Capability                      ::= CHOICE
{
    nonStandard                 NonStandardParameter,
    isdn                        CHOICE
    {
    isdnCapability              IsdnCapability,
    multilinkAdditionalConnection NULL,  -- Express this cap alone to
                                         force  association of this
                                         channel with
                                                 --an existing call
    is13871                     SEQUENCE         -"BONDING" protocol
        {
                                withIsdnCapability      IsdnCapability,
                                ...
        },

        h244                SEQUENCE                    -- channel
aggregation protocol
        {
                            withIsdnCapability          IsdnCapability,
                            ...
        },
        ...
    },
    ...
}

IsdnCapability              ::=CHOICE
{
    g711aLaw                SEQUENCE {...},
    g711uLaw                SEQUENCE {...},
    h320                    SEQUENCE {...},
    h324AnnexD              SEQUENCE {...},
    h324Multilink           SEQUENCE {...},
```

```
    group4Fax               SEQUENCE {...},
    t120                    SEQUENCE {...},
    t140                    SEQUENCE {...},     – text chatting protocol
    v110                    SEQUENCE {...},
    v120                    SEQUENCE {...},
    rfc1661                 SEQUENCE
    {
        withH323        BOOLEAN,
        ...
    },
    ...
}


{
    nonStandard                         NonStandardParameter,
    plainIsdnMode                       IsdnMode,
    h244                                IsdnMode,
    is13871                             IsdnMode,       – BONDING
protocol
    multilinkAdditionalConnection       SEQUENCE
    {
        callAssociationNumber     INTEGER (0..4294967295),
        ...
    },
    ...
}

IsdnMode                    ::= CHOICE
{
    nonStandard                 NonStandardParameter,
    g711aLaw                    SEQUENCE {...},
    g711uLaw                    SEQUENCE {...},
    h320                        SEQUENCE {...},
    h324AnnexD                  SEQUENCE {...},
    h324Multilink               SEQUENCE {...},
    group4Fax                   SEQUENCE {...},
    t120                        SEQUENCE {...},
    rfc1661                     SEQUENCE {...},
    ...
}

TerminateMessage                ::=SEQUENCE
{
    cause                       CHOICE
    {
        nonStandard             NonStandardParameter,
```

```
            timerExpiration          NULL,
            roleCollision            NULL,
            noSuitableModes          NULL,
            invalidModeSelected      NULL,
            protocolViolation        NULL,
            modeNotAvailable         NULL,
            ...
        },
        ...
}

--
=======================================================
-- Non standard Message definitions
--
=======================================================

NonStandardMessage            ::=SEQUENCE
{
    nonStandardData           NonStandardParameter,
    ...
}

NonStandardParameter          ::=SEQUENCE
{
    nonStandardIdentifier     NonStandardIdentifier,
    data                      OCTET STRING
}

NonStandardIdentifier         ::=CHOICE
{
    object                    OBJECT IDENTIFIER,
    h221NonStandard           SEQUENCE
    {
        t35CountryCode        INTEGER (0..255),-- country, per T.35
        t35Extension          INTEGER (0..255),-- assigned ationally
        manufacturerCode      INTEGER (0..65535)-- assigned
nationally
    }
}
```

Necessary UMTS (GSM) codings in the frame of reference of the patent application:

<u>Bearer Capability</u>

- *Bearer Capability Information Element*

	Transfer mode (octet 3)
0	Circuit mode (Bit 4)
	Bearer Capability Information Element
	Duplex mode (octet 4)
1	Full duplex (Bit 4)
	Bearer Capability Information Element
	Synchronous/Asynchronous (octet 6)
0	Synchronous (Bit 1)
	Bearer Capability Information Element
	Fixed network user rate (octet 6d)
00100	28.8 kbps (Bits 54321)
	Bearer Capability Information Element
	Acceptable channel codings (octet 6e)
1	TCH/F14.4 acceptable (Bit 7)
	Bearer Capability Information Element
	Acceptable channel codings (octet 6e)
1	TCH/F9.6 acceptable (Bit 5)
	Bearer Capability Information Element
	Acceptable channel codings (octet 6e)
1	TCH/F4.8 acceptable (Bit 4)
	Bearer Capability Information Element
	Maximum number of traffic channels (octet 6e)
001	2 TCH (Bits 321)
	Bearer Capability Information Element
	Connection element (octet 6c)
00	Transparent (Bits 76)
	Bearer Capability Information Element
	Other modem type (octet 6d)
10	V.34 (Bits 76)
	Bearer Capability Information Element
	User information layer 1 protocol (octet 5)
00110	Recommendations H.223 and H.245 (Bits 54321)

The latest information coding point is not present in current GSM BC information element. It is taken from Q.931 and should be added to future UMTS BC information element.

<u>High Layer Compatibility</u>

	High Layer Compatibility Information Element
	High layer characteristics identification (octet 4)
1100001	PSTN Multimedia Terminal (Bits 7654321)

The present invention includes any novel feature or combination of features disclosed herein either explicitly or any generalisation thereof irrespective of whether or not it relates to the claimed invention or mitigates any or all of the problems addressed.

In view of the foregoing description it will be evident to a person skilled in the art that various modifications may be made within the scope of the invention. For example, the reduced numbering scheme is not restriceted to only one or two numbers for call type differentiation. It will be appreciated that the reference to H.324 throughout this text is purely exemplary and that the invention is also applicable to other data types.

Claims

1. A switch for establishing a call between a terminal of an originating network and a terminal of a terminating network, the switch comprising:

an input for receiving call type information in a first format from the originating network;

means for reformatting received call type information into a second format;

an output means for outputting the call type information in the second format over the terminating network; and

connection means for completing a connection, suitable for the identified call type, between the terminals.

2. A switch as claimed in claim 1, wherein the call type information comprises teleservice information.

3. A switch as claimed in claim 1 or 2, wherein the call type information comprises bearer service information.

4. A switch as claimed in any preceding claim, wherein the first format is an in band format.

5. A switch as claimed in any preceding claim, wherein the second format is an out band format.

6. A switch as claimed in any preceding claim, wherein the terminating network is digital.

7. A switch as claimed in any preceding claim, wherein the originating network is analogue.

8. A switch as claimed in any preceding claim, wherein one of the networks is a wireless communications network.

9. A switch as claimed in claim 8, wherein the wireless communication network is a universal mobile telecommunications system (UMTS) network.

10. A switch as claimed in claim 8, wherein the wireless communication network is a GSM network.

11. A switch as claimed in claim 9 or 10, wherein the switch is a mobile switching centre.

12. A switch as claimed in any of claims 8 to 11, comprising a transcoder.

13. A switch as claimed in any of claims 8 to 12 wherein the wireless communications network is the terminating network.

14. A switch as claimed in any of claims 1 to 5, wherein one of the networks is a fixed line network.

15. A switch as claimed in claim 14, wherein the fixed line network is a PSTN network.

16. A switch as claimed in claim 14, wherein the fixed line network is an ISDN network.

17. A switch as claimed in any of claims 14 to 16, wherein the fixed line network is the originating network.

18. A switch as claimed in any preceding claim, further comprising:

means, coupled to the input, for determining primary call type information on the basis of a subscriber number, for forwarding first primary

call type information to the output, and for forwarding further primary call type information to the reformatting means.

19. A method for establishing a call between a terminal of an originating network and a terminal of a terminating network, the method comprising:

receiving call type information in a first format from the originating network;

reformatting received call type information into a second format;

outputting the call type information in the second format over the terminating network; and

completing a connection, suitable for the identified call type, between the terminals.

20. A method for establishing a call between a terminal of an originating network and a terminal of a terminating network, the method comprising:

establishing a call of a predetermined type;

transmitting call type information in a first format from the originating terminal to the terminating network;

reformatting received call type information into a second format;

transmitting the call type information in the second format to the terminating terminal; and

establishing a connection, suitable for the identified call type, between the terminals.

21. A method as claimed in claim 19 or 20, further comprising:

determining primary call type information on the basis of a subscriber number;

providing a predetermined connection if the primary call type corresponds to the call type of the predetermined connection; and

performing the reformatting, transmitting and establishing steps if the primary call type is another type.

22. A switching system for establishing a call between a terminal of an originating network and a terminal of a terminating network, the switching system comprising:

means for receiving call type information in a first format from the originating network;

means for reformatting received call type information into a second format;

means for transmitting the call type information in the second format over the terminating network; and

connection means for completing a connection, suitable for the identified call type, between the terminals.

23. A switch for establishing a call between a terminal of an originating network and a terminal of a terminating network, substantially as hereinbefore described with reference to, and/or as illustrated in any one, or any combination, of Figures 2b to 8b of the accompanying drawings.

24. A system for establishing a call between a terminal of an originating analogue network and a terminal of a digital terminating network, substantially as hereinbefore described with reference to, and/or as illustrated in any one, or any combination, of Figures 2b to 8b of the accompanying drawings.

25. A method for establishing a call between a terminal of an originating analogue network and a terminal of a digital terminating network, substantially as hereinbefore described with reference to, and/or as illustrated in any one, or any combination, of Figures 2b to 8b of the accompanying drawings.

The Patent Office

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Application No: GB 9906756.3
Claims searched: 1-25

Examiner: Richard Howe
Date of search: 10 June 1999

Patents Act 1977
Search Report under Section 17

Databases searched:

UK Patent Office collections, including GB, EP, WO & US patent specifications, in:

UK Cl (Ed.Q): H4K (KTA, KTM, KTK) ; H4P (PF)

Int Cl (Ed.6): H04L (12/46, 12/64, 12/66, 29/06)

Other: Online : wpi; epodoc; japio

Documents considered to be relevant:

Category	Identity of document and relevant passage		Relevant to claims
X	EP 0 735 724 A2	(AT&T) - see abstract	1,19,20,22 at least
X	EP 0 568 520 A2	(Fisher Controls) - see abstract	1,19,20,22 at least
X	EP 0 503 207 A1	(Interntional Business Machines) - see abstract	1,19,20,22 at least
X	EP 0 287 878 A1	(Westinghouse) - see absract	1,19,20,22 at least
X	WO 98/16046 A1	(Microsoft) - see abstract	1,19,20,22 at least
X	WO 98/07258 A1	(Ascend) - see abstract	1,19,20,22 at least
X	WO 97/48202 A1	(Epoch) - see abstract	1,19,20,22 at least
X	WO 97/47114 A1	(Cignal) - see abstract	1,19,20,22 at least
X	WO 97/16007 A1	(Telecom Finland) - see abstract	1,19,20,22 at least

X	Document indicating lack of novelty or inventive step
Y	Document indicating lack of inventive step if combined with one or more other documents of same category.
&	Member of the same patent family
A	Document indicating technological background and/or state of the art.
P	Document published on or after the declared priority date but before the filing date of this invention.
E	Patent document published on or after, but with priority date earlier than, the filing date of this application.